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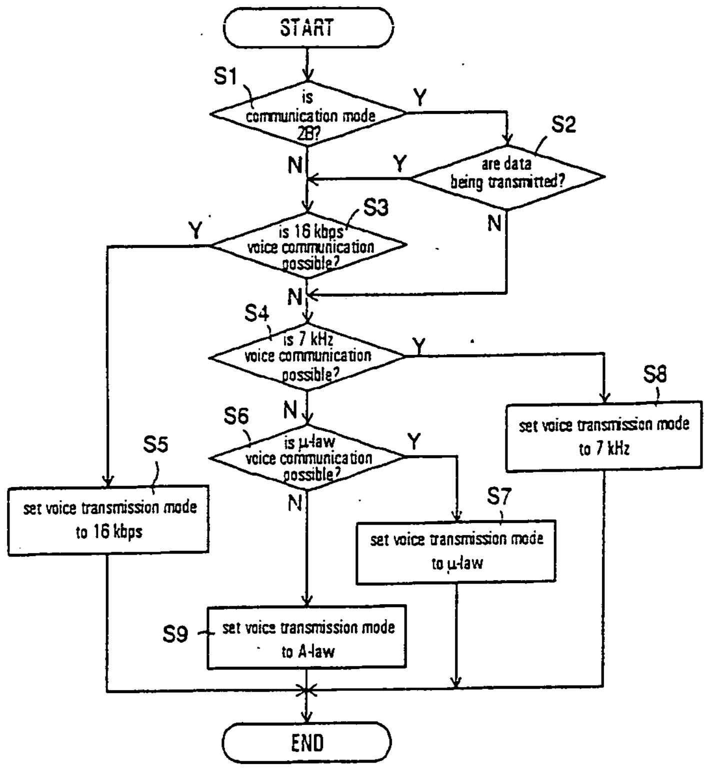
Title Terminal Equipment

Abstract

PURPOSE: To optimise voice transfer.

CONSTITUTION: If the communication mode is 2B (see S1), a check is made to ascertain whether or not data are being transmitted (S2). If the communication mode is 1B (see S1) or if data are being transmitted in 2B communication mode (see S2), the voice communication possibilities are checked in S3 and subsequent steps. If the communication mode is 2B but data are not being transmitted (see S2), the voice communication possibilities are checked in S4 and subsequent steps. When the information transfer rate in a communication channel is 1B or when low-speed data are being transmitted [1]*, the mode employed for voice transmission is selected in the following order of priority: 16 kbps (S5), 7 kHz (S8), μ-law (S7) and A-law (S9). Otherwise, the mode employed for voice transmission is selected in the order: 7 kHz (S8), μ-law (S7) and A-law (S9). This serves to optimise the voice transmission mode. [2]



Claims

1. Terminal equipment having a plurality of voice codecs, said terminal equipment characterised in that it comprises a means for detecting the transfer rate of the communication channel, a means for switching among said plurality of voice codecs, and a means for detecting a voice codec that can be selected from said plurality of voice codecs; and further characterised in that it selects an optimum voice codec in accordance with the transfer rate of the communication channel.

2. The terminal equipment according to Claim 1, wherein there is provided a means for detecting if data are being transferred, and wherein an optimum voice codec is

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* Numbers in square brackets refer to Translator's Notes appended to the translation.

selected in accordance with the transfer rate of the communication channel and in accordance with whether data are being transferred.

3. The terminal equipment according to Claim 1, wherein, when selecting a voice codec, if there are voice codecs with similar transfer rates, said terminal equipment switches to a high sound quality voice codec. [3]

4. The terminal equipment according to Claim 1, said terminal equipment switching the voice codec when the communication state changes.

Detailed Description of the Invention

Industrial field of utilisation

(1) The present invention relates to terminal equipment, and more particularly relates to terminal equipment for communicating, via a communication channel, combined information comprising voice, images [4], data, etc.*

Prior art

(2) A combined image and voice service such as video telephony or video-conferencing using digital channels has been attracting attention. Service specifications, protocol specifications and multimedia multiplexed frame structure specifications for implementing such services have been published as the H-series of recommendations (previously CCITT recommendations).

(3) These recommendations specify the procedures for setting up physical end-to-end connections and using the in-channel to establish synchronisation. They also specify capability switching sequences for negotiating common functions with a remote terminal using a BAS [5] in the in-channel, and mode switching sequences, these latter depending on the specification of the communication modes. Communication of combined information such as images, voice and data can be carried out between terminals by means of procedures conforming to these recommendations.

(4) The communication mode is determined by the transmitting side comparing the receiving capability of the called party with its own transmitting capability, and notifying the receiving side of common functions. However, within the range of capabilities to which a terminal can change or switch its own capability, which communication mode to use is outside the scope of the specifications.

(5) In the communication of combined information consisting of images, voice and data, a voice coding scheme for the voice information is specified first of all, and next a transfer rate for data is specified. The remainder left after subtracting the transfer rate of the voice information and the transfer rate of the data from the total transfer rate of the communication channel is allocated to the transfer of the image information.

(6) A plurality of voice codecs can be provided in accordance with voice transfer rate and audio quality. To guarantee compatibility with existing telephony, it is essential to implement either the PCM A-law codec or the PCM μ-law codec.

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* Numbers in round brackets at the beginning of paragraphs correspond to the paragraph numbering in the Japanese patent document.

Problem that the invention will overcome

(7) A problem that has been encountered with low transfer rate communication channels is that when the proportion of bandwidth taken up by the transfer of voice and data is high, the transfer rate of image information becomes comparatively low, leading to decreased frame rate and deterioration of image quality.

(8) It is an object of the present invention to provide terminal equipment whereby this adverse state of affairs does not occur.

Means for overcoming problem

(9) Terminal equipment according to the present invention has a plurality of voice codecs and is characterised in that it comprises a means for detecting the transfer rate of the communication channel, a means for switching among said plurality of voice codecs, and a means for detecting a voice codec that can be selected from said plurality of voice codecs; and further characterised in that it selects an optimum voice codec in accordance with the transfer rate of the communication channel.

Working of the invention

(10) The various means described above enable the voice codec which is used to be switched to an appropriate one in accordance with a change in the communication conditions. This ensures that the transfer rate allocated to the transfer of image information does not become excessively low.

Embodiment

(11) An embodiment of the present invention will now be described with reference to the accompanying drawings.

(12) FIG. 1 is a schematic block diagram of terminal equipment for image, voice and data communication and depicts an embodiment of the present invention.

(13) The following numerals in FIG. 1 reference the following elements: 10... camera for imaging a user; 12... image monitor for displaying the user's image, a received image, a control screen, etc.; 14... image coding and decoding circuit for coding an image signal to be transmitted and for decoding received coded image data, in accordance with recommendation H.261 (formerly a CCITT recommendation); 16... image interface for performing NTSC/CIF conversion on the output of camera 10 and supplying the converted output to image coding and decoding circuit 14, and for performing CIF/NTSC conversion on the received image that has been decoded by image coding and decoding circuit 14 and for outputting the converted image to monitor 12. Image interface 16 is also provided with functions that enable it to select between the image input by camera 10 and the received image, and to deliver combined and split displays.

(14) Referring again to FIG. 1, the following numerals reference the following elements: 18... handset for voice telephony; 20... microphone for voice input; 22... speaker for voice output; 24... voice coding and decoding circuit (voice codec) for coding a voice signal to be transmitted and for decoding a received coded voice signal. Numeral 26 references a voice interface between handset 18, microphone 20 and speaker 22 on the one hand, and voice coding and decoding circuit 24 on the other.

(15) Voice interface 26 performs the following processing: on/off hook detection for detecting whether handset 18 is on-hook or off-hook; echo cancelling for eliminating echo when using microphone 20 and speaker 22; and generation of various tones such as dial tone, audible ringing signal, busy tone and ringtone.

(16) Numeral 28 in FIG. 1 references a data terminal for input and output of data, and for displaying and processing data. Numeral 30 references a data interface serving data terminal 28.

(17) Numeral 32 in FIG. 1 references a system control circuit for controlling the overall system, this control circuit comprising well-known devices such as a CPU, ROM, RAM and auxiliary storage devices. Numeral 34 references an operator controlled device (e.g., a numeric keypad, a keyboard or a touch panel) whereby a user inputs prescribed instructions to system control circuit 32.

(18) Numeral 36 in FIG. 1 references a channel interface for controlling channels. This channel interface conforms to the ISDN User-Network Interface Protocol. Numeral 38 references a multiplexing and demultiplexing circuit for demultiplexing information received from channel interface 36 and supplying it, according to its content, to image coding and decoding circuit 14, voice coding and decoding circuit 24, data interface 30 and system control circuit 32; and for multiplexing coded image data from image coding and decoding circuit 14, coded voice data from voice coding and decoding circuit 24, data from data interface 30 and control commands from system control circuit 32, into transmission frame units, and supplying these to channel interface 36.

(19) FIG. 2 is a schematic block diagram of the internal constitution of voice coding and decoding circuit 24. The following numerals in FIG. 2 reference the following elements: 40... 64 kbps PCM A-law codec, 42... 64 kbps PCM μ-law codec, 44... 16 kbps (e.g. APC-AB) codec, 46... 48 kbps SB-ADPCM codec supporting 7 kHz [6], 48... codec switching circuit for selecting, from among 40, 42, 44 and 46 and in accordance with control commands from system control circuit 32, a codec to be used for transmitting and receiving.

(20) FIG. 3 shows the internal constitution (implemented in software or hardware) of system control circuit 32. In FIG. 3, numeral 50 references a main control portion for controlling the devices constituting the terminal device and for communicating with each of these constituent devices. Numeral 52 references a D-channel control portion for performing D-channel control of for example call setup and disconnect. Numeral 54 references an in-channel control portion for performing in-channel synchronisation, capability switching, mode switching, etc.

(21) The working of this embodiment will now be described with reference to FIG. 4 and FIG. 5, of which FIG. 4 is a flowchart of the algorithm whereby system control circuit 32 determines the voice transmission mode.

(22) Firstly, the information transfer rate of the communication channel is checked (S1). The present embodiment presupposes an ISDN Basic Rate Interface terminal and hence the information transfer rate is 1B or 2B. If the communication mode is 2B (see S1), a check is made to ascertain whether or not data are being transmitted (S2).

(23) If the communication mode is 1B (see S1) or if data are being transmitted in 2B communication mode (see S2), the voice communication possibilities are checked in S3 and subsequent steps. If the communication mode is 2B but data are not being transmitted, the voice communication possibilities are checked in S4 and subsequent steps. It may be noted that communication capabilities are common to the called terminal and the local terminal.

(24) If voice communication is possible at 16 kbps (see S3), the voice transmission mode is set to 16 kbps (S5). If voice communication is not possible at 16 kbps (see S3), a check is made to ascertain whether or not there is capability for 7 kHz voice (see S4). If such communication is possible, the voice transmission mode is set to 7 kHz (S8).

(25) If 7 kHz voice communication is not possible (see S4), a check is made to ascertain whether there is capability for µ-law voice (see S6). If there is, the voice transmission mode is set to µ-law (S7). If there is no µ-law voice capability (see S6), the voice transmission mode is set to A-law (S9).

(26) As a result of the foregoing control, when the information transfer rate in a communication channel is 1B or when low-speed data are being transmitted, the mode employed for voice transmission is selected in the following order of priority: 16 kbps, 7 kHz, µ-law and A-law. Otherwise, it is selected in the order: 7 kHz, µ-law and A-law. This serves to optimise the voice transmission mode.

(27) FIG. 5 is a flowchart showing at which junctures during a call the optimisation of the voice transmission mode shown in FIG. 4 is performed.

(28) Firstly, a check is made to ascertain whether or not communication is in progress (S11). If communication is not in progress, processing is terminated. If communication is in progress, a check is made to ascertain whether or not there is any change in communication capability (S12). A change in communication capability occurs when the local terminal capability has been changed or when the remote terminal capability has been changed. If communication capability has been changed, the voice transmission mode is optimised (S15).

(29) If there is no change in communication capability, the voice transmission mode is optimised (S15) when data transmission begins or ends (S13).

(30) The voice transmission mode is also optimised (S15) when there has been a change in the information transfer rate in a communication channel (S14).

Effect of the invention

(31) As will readily be understood from the foregoing description, according to the present invention an optimum voice codec is selected in accordance with transfer rate in a communication channel and in accordance with whether or not data are being transferred. As a result, deterioration of picture quality can be decreased even when there are fluctuations in the transfer rate of the communication channel or in the data transfer rate.

Brief Description of the Drawings

FIG. 1 is a schematic block diagram of an embodiment of the present invention.

FIG. 2 is a schematic block diagram of the internal constitution of voice coding and decoding circuit 24.

FIG. 3 is a schematic block diagram of the internal constitution of system control circuit 32.

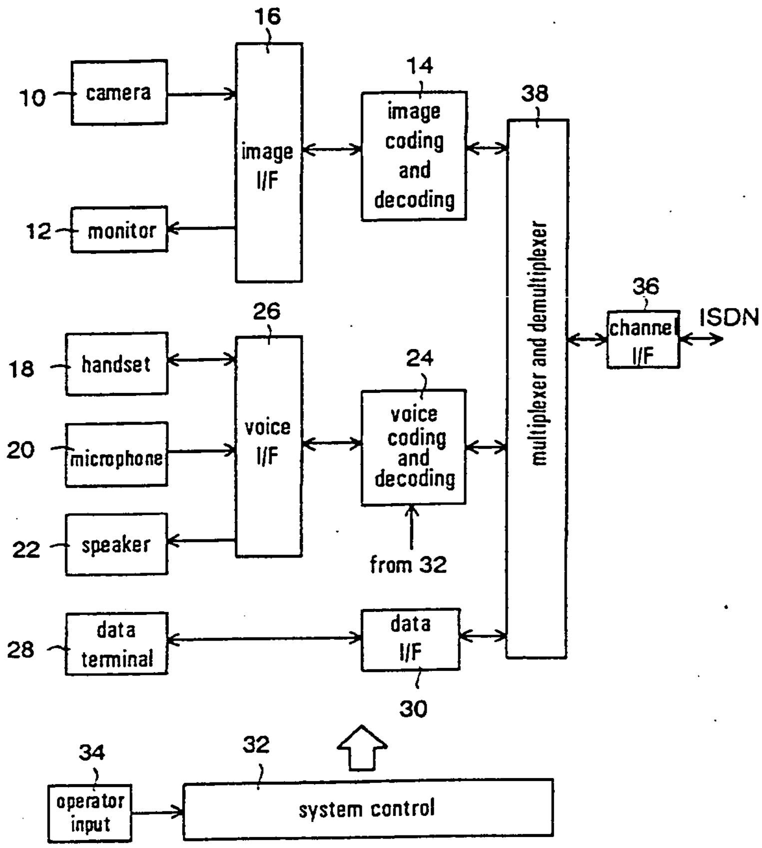
FIG. 4 is a flowchart showing the algorithm employed for optimisation of voice transmission mode in the present embodiment.

FIG. 5 is a flowchart showing the junctures at which the optimisation of voice transmission mode is performed.

Key to referencing numerals

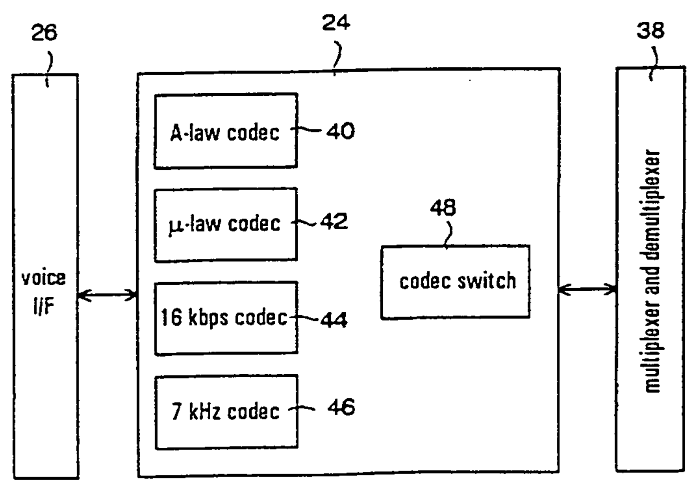
10..........camera
12..........image monitor
14..........image coding and decoding circuit
16..........image interface
18..........handset
20..........microphone
22..........speaker
24..........voice coding and decoding circuit
26..........voice interface
28..........data terminal
30..........data interface
32..........system control circuit
34..........operator controlled device
36..........channel interface
38..........multiplexing and demultiplexing circuit
40..........64 kbps PCM A-law codec
42..........64 kbps PCM μ-law codec
44..........16 kbps (e.g., APC-AB) codec
46..........48 kbps SB-ADPCM codec supporting 7 kHz
48..........codec switching circuit
50..........main control portion
52..........D-channel control portion
54..........in-channel control portion

FIG. 1



16
14
38
10
camera
image I/F
image coding and decoding
12
monitor
multiplexer and demultiplexer
36
channel I/F
ISDN
26
18
handset
24
voice I/F
voice coding and decoding
20
microphone
22
speaker
from 32
28
data terminal
data I/F
30
34
32
operator input
system control

FIG. 2



26
24
38
A-law codec
40
μ-law codec
42
48
voice I/F
codec switch
16 kbps codec
44
7 kHz codec
46
multiplexer and demultiplexer

FIG.3

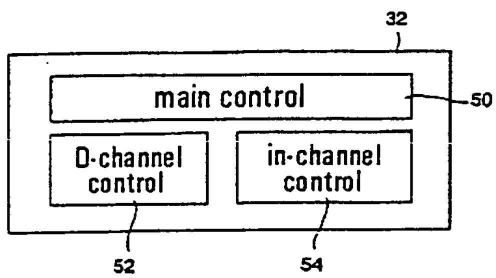

32
main control
50
D-channel control
in-channel control
52
54

FIG. 4

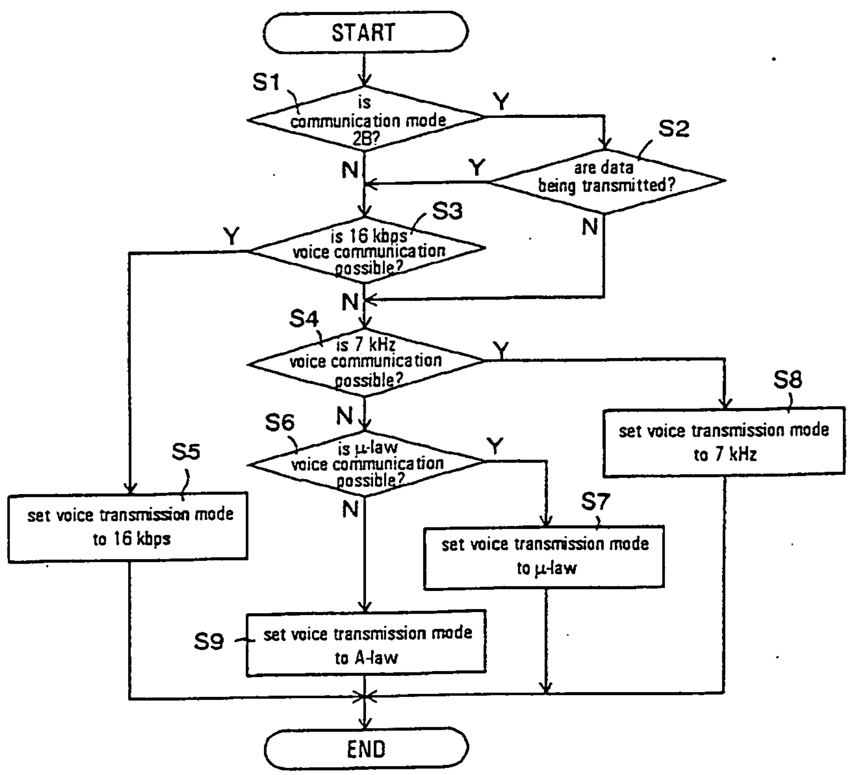
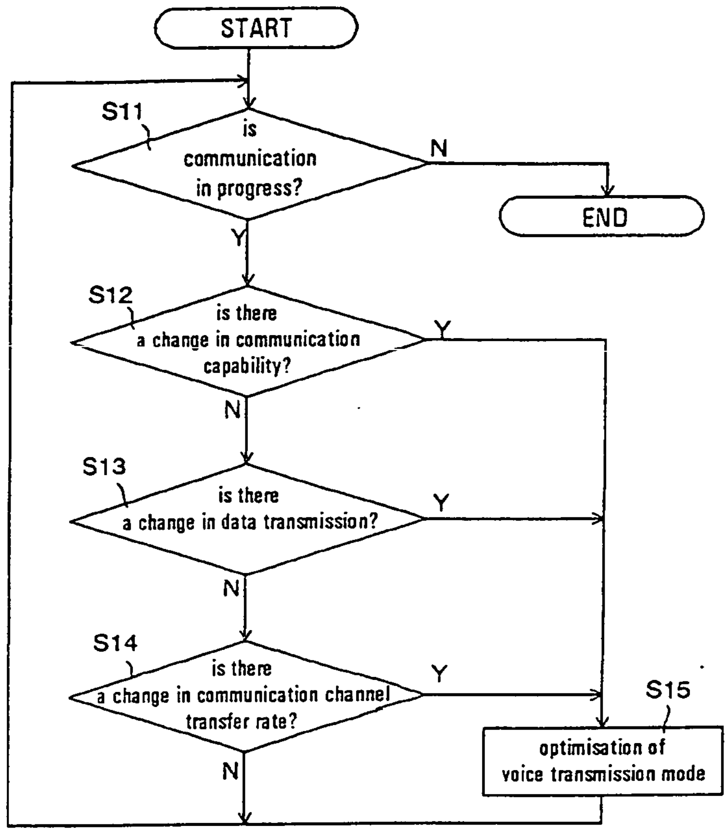

START
S1
is communication mode 2B?
Y
S2
are data being transmitted?
N
Y
S3
is 16 kbps voice communication possible?
Y
N
N
S4
is 7 kHz voice communication possible?
Y
S8
set voice transmission mode to 7 kHz
N
S6
is μ-law voice communication possible?
Y
S5
set voice transmission mode to 16 kbps
N
S7
set voice transmission mode to μ-law
S9
set voice transmission mode to A-law
END

FIG. 5



TRANSLATOR'S NOTES

1. The Japanese that I have translated as "when low-speed data are being transmitted" is literally "during LSD transmission".
2. The Japanese term that I have translated, throughout this document, as "voice" can also signify simply "audio".
3. Sic. The Japanese does not explicitly state that under these circumstances the terminal equipment switches to the codec offering the ***higher*** sound quality or the ***highest*** sound quality.
4. The Japanese term that I have translated as "images" also has the narrower meaning of "video".
5. "BAS" probably refers to a bit-rate allocation signal.
6. The writer presumably means that this SB-ADPCM codec supports 7 kHz audio at 48 kbps.